

**Apparatus for analyzing an audio signal with regard to
rhythm information of the audio signal by using an
autocorrelation function**

5 Cross-Reference to Related Application:

This application is a continuation of copending
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published in English.

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1. Field of the Invention:

The present invention relates to signal processing concepts
and particularly to the analysis of audio signals with
15 regard to rhythm information.

2. Description of the related art:

Over the last years, the availability of multimedia data
20 material, such as audio or video data, has increased
significantly. This is due to a series of technical
factors, based particularly on the broad availability of
the internet, of efficient computer hardware and software
as well as efficient methods for data compression, i.e.
25 source encoding of audio and video methods.

The huge amount of audio visual data, that are available
worldwide, for example on the internet, require concepts,
which make it possible, to be able to touch, catagolize,
30 etc. these data according to content criteria. There is a
demand to be able to search for and find multimedia data in
a calculated way by specifying useful criteria.

This requires so-called "content-based" techniques, which
35 extract so-called features from the audiovisual data, which
represent important characteristic properties of the
signal. Based on such features and combination of these
features, respectively, similarity relations and common

features, respectively, between audio or video signals can be derived. This is performed by comparing and relating, respectively, the extracted feature values from the different signals, which are also simply referred to as
5 "pieces".

The determination and extraction, respectively, of features that do not only have signal-theoretical but immediate semantic meaning, i.e. represent properties immediately
10 received by the listener, is of special interest.

This enables the user to phrase search requests in a simple and intuitive way to find pieces from the whole existing data inventory of an audio signal data bank. In the same
15 way, semantically relevant features permit to model similarity relationships between pieces, which come close to the human perception. The usage of features, which have semantic meaning, enables also, for example, an automatic proposal of pieces of interest for a user, if his
20 preferences are known.

In the area of music analysis, the tempo is an important musical parameter, which has semantic meaning. The tempo is usually measured in beats per minute (BPM). The automatic
25 extraction of the tempo as well as of the bar emphasis of the "beat", or generally the automatic extraction of rhythm information, respectively, is an example for capturing a semantically important feature of a piece of music.

30 Further, there is a demand that the extraction of features, i.e. extracting rhythm information from an audio signal, can take place in a robust and computing-efficient way. Robustness means that it does not matter whether the piece has been source-encoded and decoded again, whether the
35 piece is played via a loudspeaker and received from a microphone, whether it is played loud or soft, or whether it is played by one instrument or by a plurality of instruments.

For determining the bar emphasis and thereby also the tempo, i.e. for determining rhythm information, the term "beat tracking" has been established among the experts. It is known from the prior art to perform beat tracking based on note-like and transcribed, respectively, signal representation, i.e. in midi format. However, it is the aim not to need such metarepresentations, but to perform an analysis directly with, for example, a PCM-encoded or, generally, a digitally present audio signal.

The expert publication "Tempo and Beat Analysis of Acoustic Musical Signals" by Eric D. Scheirer, J. Acoust. Soc. Am. 103:1, (Jan 1998) pp. 588 - 601 discloses a method for automatical extraction of a rhythmical pulse from musical extracts. The input signal is split up in a series of sub-bands via a filter bank, for example in 6 sub-bands with transition frequencies of 200 Hz, 400 Hz, 800 Hz, 1600 Hz and 3200 Hz. Low pass filtering is performed for the first sub-band. High-pass filtering is performed for the last sub-band, bandpass filtering is described for the other intermediate sub-bands. Every sub-band is processed as follows. First, the sub-band signal is rectified. Put another way, the absolute value of the samples is determined. The resulting n values will then be smoothed, for example by averaging over an appropriate window, to obtain an envelope signal. For decreasing the computing complexity, the envelope signal can be sub-sampled. The envelope signals will be differentiated, i.e. sudden changes of the signal amplitude will be passed on preferably by the differentiating filter. The result is then limited to non-negative values. Every envelope signal will then be put in a bank of resonant filters, i.e. oscillators, which each comprise a filter for every tempo region, so that the filter matching the musical tempo is excited the most. The energy of the output signal is calculated for every filter as measure for matching the tempo of the input signal to the tempo belonging to the

filter. The energies for every tempo will then be summed over all sub-bands, wherein the largest energy sum characterizes the tempo supplied as a result, i.e. the rhythm information. Contrary to auto correlation functions,
5 it is advantageous that the oscillator bank reacts to a stimulus also with output signals at double, triple, etc. the tempo or also at rational multiples (such as $2/3$, $3/4$ of the tempo. An auto correlation function does not have that property, it provides only output signals at one half,
10 one third, etc. of the tempo.

A significant disadvantage of this method is the large computing and memory complexity, particularly for the realization of the large number of oscillators resonating
15 in parallel, only one of which is finally chosen. This makes an efficient implementation, such as for real-time applications, almost impossible.

The expert publication "Pulse Tracking with a Pitch Tracker" by Eric D. Scheirer, Proc. 1997 Workshop on Applications of Signal Processing to Audio and Acoustics, Mohonk, NY, Oct 1997 describes a comparison of the above-described oscillator concept to an alternative concept,
20 which is based on the use of autocorrelation functions for the extraction of the periodicity from an audio signal, i.e. the rhythm information of a signal. An algorithm for the modulation of the human pitch perception is used for beat tracking.
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The known algorithm is illustrated in Fig. 3 as a block diagram. The audio signal is fed into an analysis filterbank 302 via the audio input 300. The analysis filterbank generates a number n of channels, i.e. of individual sub-band signals, from the audio input. Every
30 sub-band signal contains a certain area of frequencies of the audio signal. The filters of the analysis filterbank are chosen such that they approximate the selection
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characteristic of the human inner ear. Such an analysis filterbank is also referred to as gamma tone filterbank.

5 The rhythm information of every sub-band is evaluated in means 304a to 304c. For every input signal, first, an envelope-like output signal is calculated (with regard to a so-called inner hair cell processing in the ear) and sub-sampled. From this result, an autocorrelation function (ACF) is calculated, to obtain the periodicity of the
10 signal as a function of the lag.

At the output of means 304a to 304c, an autocorrelation function is present for every sub-band signal, which represents the rhythm information of every sub-band signal.
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The individual autocorrelation functions of the sub-band signals will then be combined in means 306 by summation, to obtain a sum autocorrelation function (SACF), which reproduces the rhythm information of the signal at the
20 audio input 300. This information can be output at a tempo output 308. High values in the sum autocorrelation show that a high periodicity of the note beginnings is present for a lag associated to a peak of the SACF. Thus, for example the highest value of the sum autocorrelation
25 function is searched for within the musically useful lags.

Musically useful lags are, for example, the tempo range between 60 bpm and 200 bpm. Means 306 can further be disposed to transform a lag time into tempo information.
30 Thus, a peak of a lag of one second corresponds, for example, a tempo of 60 beats per minute. Smaller lags indicate higher tempos, while higher lags indicate smaller tempos than 60 bpm.

35 This method has an advantage compared to the first mentioned method, since no oscillators have to be implemented with a high computing and storage effort. On the other hand, the concept is disadvantageous in that the

quality of the results depends strongly on the type of the audio signal. If, for example, a dominant rhythm instrument can be heard from an audio signal, the concept described in Fig. 3 will work well. If, however, the voice is dominant, which will provide no particularly clear rhythm information, the rhythm determination will be ambiguous. However, a band could be present in the audio signal, which merely contains rhythm information, such as a higher frequency band, where, for example, a Hi-hat of drums is positioned, or a lower frequency band, where the large drum of the drums is positioned on the frequency scale. Due to the combination of individual information, the fairly clear information of these particular sub-bands is superimposed and "diluted", respectively, by the ambiguous information of the other sub-bands.

Another problem when using autocorrelation functions for extracting the periodicity of a sub-band signal is that the sum autocorrelation function, which is obtained by means of output 306, is ambiguous. The sum autocorrelation function at output 306 is ambiguous in that an autocorrelation function peak is also generated at a plurality of a lag. This is understandable by the fact that the sinus component with a period of t_0 , when subjected to an autocorrelation function processing, generates, apart from the wanted maximum at t_0 , also maxima at the plurality of the lags, i.e. at $2t_0$, $3t_0$, etc.

The expert publication "A Computationally Efficient Multipitch Analysis Model " by Tolonen and Karjalainen, IEEE Transactions on Speech and Audio Processing, Vol. 8, Nov 2000, discloses a computing time-efficient model for a periodicity analysis of complex audio signals. The calculating model divides the signal into two channels, into a channel below 1000 Hz and into a channel above 1000 Hz. There from, an autocorrelation of the lower channel and an autocorrelation of the envelope of the upper channel are calculated. Finally, the two autocorrelation

functions will be summed. In order to eliminate the ambiguities of the sum autocorrelation function, the sum autocorrelation function is processed further, to obtain a so-called enhanced summary autocorrelation function

5 (ESACF). This post-processing of the sum autocorrelation function comprises a repeated subtraction of versions of the autocorrelation function spread with integer factors from the sum autocorrelation function with a subsequent limitation to non-negative values.

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It is a disadvantage of this concept that the ambiguities per sub-band obtained by the auto correlation function in the sub-bands are only eliminated in the sum auto correlation function but not immediately where they occur, namely in the individual sub-bands.

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A further disadvantage of this concept is the fact that the auto correlation function itself does not provide any hint to the double, triple,... of the tempo, to which an auto correlation peak is associated.

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SUMMARY OF THE INVENTION

It is the object of the present invention to provide an apparatus and a method for analyzing an audio signal with regard to rhythm information by using an auto correlation function, which is robust and computing-time-efficient.

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In accordance with a first aspect of the invention, this object is achieved by an apparatus for analyzing an audio signal with regard to rhythm information of the audio signal by using an autocorrelation function, comprising: means for dividing the audio signal into at least two sub-band signals; means for examining at least one sub-band signal with regard to a periodicity in the at least one sub-band signal by an autocorrelation function, to obtain rhythm raw-information for the sub-band signal, wherein a delay is associated to a peak of the autocorrelation

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function; means for postprocessing the rhythm raw-information for the sub-band signal determined by the autocorrelation function, to obtain postprocessed rhythm raw-information for the sub-band signal, so that in the
5 postprocessed rhythm raw-information an ambiguity in an integer plurality of a delay, to which an autocorrelation function peak is associated, is reduced, or a signal portion is added at an integer fraction of a delay, to which an autocorrelation function peak is associated; and
10 means for establishing the rhythm information of the audio signal by using the postprocessed rhythm raw-information of the sub-band signal and by using another sub-band signal of the at least two sub-band signals.

15 In accordance with a second aspect of the invention, this aspect is achieved by an apparatus for analyzing an audio signal with regard to rhythm information of the audio signal by using an autocorrelation function, comprising:
means for examining the audio signal with regard to a
20 periodicity in the audio signal, to obtain rhythm raw-information for the audio signal, wherein a delay is associated to a peak of the autocorrelation function; means for postprocessing the rhythm raw-information for the audio signal determined by the autocorrelation function, to
25 obtain postprocessed rhythm raw-information for the audio signal, so that in the postprocessed rhythm raw-information a signal portion is added at an integer fraction of a delay, to which an autocorrelation function peak is associated; and means for establishing rhythm information
30 of the audio signal by using the postprocessed rhythm raw-information of the audio signal.

In accordance with a third aspect of the invention, this object is achieved by an apparatus for analyzing an audio
35 signal with regard to rhythm information of the audio signal by using an autocorrelation function, comprising:
means for examining the audio signal with regard to a periodicity in the audio signal, to obtain rhythm raw-

information for the audio signal, wherein a delay is associated to a peak of the autocorrelation function; means for postprocessing the rhythm raw-information for the audio signal determined by the autocorrelation function, to
5 obtain postprocessed rhythm raw-information for the audio signal, by subtracting a version of the rhythm raw-information weighted by a factor unequal one and spread by an integer factor larger than one; and means for establishing the rhythm information of the audio signal by
10 using the postprocessed rhythm raw-information of the audio signal.

In accordance with a fourth aspect of the invention, this object is achieved by a method for analyzing an audio
15 signal with regard to rhythm information of the audio signal by using an autocorrelation function, comprising: dividing the audio signal into at least two sub-band signals, examining at least one sub-band signal with regard to a periodicity in the at least one sub-band signal by an autocorrelation function, to obtain rhythm raw-information
20 for the sub-band signal, wherein a delay is associated to a peak of the autocorrelation function; postprocessing the rhythm raw-information for the sub-band signal determined by the autocorrelation function, to obtain postprocessed
25 rhythm raw-information for the sub-band signal, so that in the postprocessed rhythm raw-information an ambiguity in the integer plurality of a delay, to which an autocorrelation function peak is associated, is reduced, or a signal portion is added at an integer fraction of a
30 delay, to which an autocorrelation function peak is associated; and establishing the rhythm information of the audio signal by using the postprocessed rhythm raw-information of the sub-band signal and by using a further sub-band signal of the at least two sub-band signals.

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In accordance with a fifth aspect of the invention, this object is achieved by a method for analyzing an audio signal with regard to rhythm information of the audio

signal by using an autocorrelation function, comprising:
examining the audio signal with regard to a periodicity in
the audio signal, to obtain rhythm raw-information for the
audio signal, wherein a delay is associated to a peak of
5 the autocorrelation function; postprocessing the rhythm
raw-information for the audio signal by the autocorrelation
function, to obtain postprocessed rhythm raw-information
for the audio signal, so that in the postprocessed rhythm
raw-information a signal portion is added at an integer
10 fraction of a delay, to which an autocorrelation function
peak is associated; and establishing the rhythm information
of the audio signal by using the postprocessed rhythm raw-
information of the audio signal.

15 In accordance to a sixth aspect of the invention, this
aspect is achieved by a method for analyzing an audio
signal with regard to rhythm information of the audio
signal by using an autocorrelation function, comprising:
examining the audio signal with regard to a periodicity in
20 the audio signal, to obtain rhythm raw-information for the
audio signal, wherein a delay is associated to a peak of
the autocorrelation function; postprocessing the rhythm
raw-information for the audio signal determined by the
autocorrelation function, to obtain postprocessed rhythm
25 raw-information for the audio signal, by subtracting a
version of the rhythm raw-information weighted with a
factor unequal one and spread by an integer factor larger
than one; and establishing the rhythm information of the
audio signal by using the postprocessed rhythm raw-
30 information of the audio signal.

The present invention is based on the knowledge that a
postprocessing of an autocorrelation function can be
performed sub-band-wise, to eliminate the ambiguities of
35 the autocorrelation function for periodical signals, and
tempo information, which an autocorrelation processing does
not provide, respectively, are added to the information
obtained by an autocorrelation function. According to an

aspect of the present invention, an autocorrelation function postprocessing of the sub-band signals is used to eliminate the ambiguities already "at the root", and to add "missing" rhythm information, respectively.

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According to another aspect of the present invention, postprocessing of the sum autocorrelation function is performed, to obtain postprocessed rhythm raw-information for the audio signal, so that in the postprocessed rhythm raw-information a signal part is added at an integer
10 fraction of a delay, to which an autocorrelation function peak is associated. Thereby, it is possible to generate the rhythm information not obtained by an autocorrelation function in double, triple, etc. tempi and in rational
15 pluralities, respectively, by calculating versions of the autocorrelation function compressed by an integer factor or by a rational factor, and by adding these versions to the original autocorrelation function. Contrary to the prior art, where an expensive oscillator bank is required
20 therefore, according to the invention, this takes place with weighting and addition routines, which are easy to implement.

According to another aspect of the present invention, the
25 sum autocorrelation function is further post-processed by subtracting a version of the rhythm raw-information to the autocorrelation function, which is weighted by a factor larger than zero and smaller than one, and spread by an integer factor larger than one. This has the advantage of
30 eliminating the ACF ambiguities in the integer multiple of the delay, to which an autocorrelation peak is associated. While in the prior art no weighting of the spread versions of the autocorrelation function is performed prior to subtraction, and an elimination of the ambiguities is
35 therefore only obtained in the theoretical optimum case, where the rhythm repeats itself ideally cyclically, the weighted subtraction provides the possibility to take rhythm information into account, which does not repeat

itself ideally cyclically, by an appropriate choice of weighting factors, which can, for example, take place empirically.

5 According to a preferred embodiment of the present invention, an autocorrelation function postprocessing is performed, by combining the rhythm information determined by an autocorrelation function with compressed and/or spread versions of it. In the case of using the spread
10 versions of the rhythm information, the spread versions are subtracted from the rhythm raw-information, while in the case of versions of the autocorrelation function compressed by integer factors, these compressed versions are added to the rhythm raw-information.

15 In a preferred embodiment of the invention, the compressed/spread version is weighted with a factor between zero and one prior to adding and subtracting.

20 According to another preferred embodiment of the present invention, a quality evaluation of the rhythm information is performed based on the post-processed rhythm raw-information to obtain a significance measure, such that the quality evaluation is no longer influenced by
25 autocorrelation artifacts. Thus, a secure quality evaluation becomes possible, whereby the robustness of determining rhythm information of the audio signal can be increased further.

30 Alternatively, the quality evaluation can already take place prior to the ACF postprocessing. This has the advantage that, when a flat course of the rhythm raw-information is determined, i.e. no distinct rhythm information, an ACF postprocessing for the sub-band signal
35 can be omitted, since this sub-band will anyway have no importance due to its hardly expressive rhythm information when determining rhythm information of the audio signal. In

this way, the computing and memory effort can be reduced further.

In the individual frequency bands, i.e. the sub-bands,
5 there are often differently favorable conditions for
finding rhythmical periodicities. While, for example, in
pop music often the area of the middle, such as around
1 kHz, the signal is dominated by a voice not corresponding
to the beat, in the higher frequency areas, often mainly
10 percussion sounds are present, such as the hihat of the
drums, which allow a very good extraction of rhythmical
regularities. In other words, different frequency bands
contain a different amount of rhythmical information,
depending on the audio signal, and have a different quality
15 or significance for the rhythm information of the audio
signal, respectively.

Therefore, according to the invention, the audio signal is
first divided into sub-band signals. Every sub-band signal
20 is examined with regard to its periodicity, to obtain
rhythm raw-information for every sub-band signal.
Thereupon, according to the present invention, an
evaluation of the quality of the periodicity of every sub-
band signal is performed to obtain a significance measure
25 for every sub-band signal. A high significance measure
indicates that clear rhythm information is present in this
sub-band signal, while a low significance measure indicates
that less clear rhythm information is present in this sub-
band signal.

30 According to a preferred embodiment of the present
invention, when examining a sub-band signal with regard to
its periodicity, first, a modified envelope of the sub-band
signal is calculated, and then an autocorrelation function
35 of the envelope is calculated. The autocorrelation function
of the envelope represents the rhythm raw-information.
Clear rhythm information is present when the
autocorrelation function shows clear maxima, while less

clear rhythm information is present when the autocorrelation function of the envelope of the sub-band signal has less significant signal peaks or no signal peaks at all. An autocorrelation function, which has clear signal peaks, will thus obtain a high significance measure, while an autocorrelation function, which has a relatively flat signal form, will obtain a low significance measure. As discussed above, the artefacts of the autocorrelation functions will be eliminated according to the invention.

The individual rhythm raw-information of the individual sub-band signal are not combined only "blindly", but under consideration of the significance measure for every sub-band signal to obtain the rhythm information of the audio signal. If a sub-band signal has a high significance measure, it is preferred when establishing the rhythm information, while a sub-band signal, which has a low significance measure, i.e., which has a low quality with regard to the rhythm information, is hardly or, in the extreme case, not considered at all when establishing the rhythm information of the audio signal.

This can be implemented computing-time-efficiently in a good way by a weighting factor, which depends on the significance measure. While a sub-band signal, which has a good quality for the rhythm information, i.e., which has a high significance measure, could obtain a weighting factor of 1, another sub-band signal, which has a smaller significance measure, will obtain a weighting factor smaller than 1. In the extreme case, a sub-band signal, which has a totally flat autocorrelation function, will have a weighting factor of 0. The weighted autocorrelation functions, i.e. the weighted rhythm raw-information, will then simply be summed up. When merely one sub-band signal of all sub-band signals supplies good rhythm information, while the other sub-band signals have autocorrelation functions with a flat signal form, this weighting can, in the extreme case, lead to the fact that all sub-band

signals apart from the one sub-band signal obtain a weighting factor of 0, i.e. are not considered at all when establishing the rhythm information, so that the rhythm information of the audio signal are merely established from one single sub-band signal.

The inventive concept is advantageous in that it enables a robust determination of the rhythm information, since sub-band signals with no clear and even differing rhythm information, respectively, i.e. when the voice has a different rhythm than the actual beat of the piece, do not dilute and "corrupt" the rhythm information of the audio signal, respectively. Above that, very noise-like sub-band signals, which provide a system autocorrelation function with a totally flat signal form, will not decrease the signal noise ratio when determining the rhythm information. Exactly this would occur, however, when, as in the prior art, simply all autocorrelation functions of the sub-band signals with the same weight are summed up.

It is another advantage of the inventive method, that a significance measure can be determined with small additional computing effort, and that the evaluation of the rhythm raw-information with the significance measure and the following summing can be performed efficiently without large storage and computing-time effort, which recommends the inventive concept particularly also for real-time applications.

Brief Description of the Drawings

Preferred embodiments of the present invention will be discussed in more detail below with reference to the accompanying drawings in which:

Fig. 1 a block diagram of an apparatus for analyzing an audio signal with a quality evaluation of the rhythm raw-information;

- Fig. 2 a block diagram of an apparatus for analyzing an audio signal by using weighting factors based on the significance measures;
- 5
- Fig. 3 a block diagram of a known apparatus for analyzing an audio signal with regard to rhythm information;
- Fig. 4 a block diagram of an apparatus for analyzing an audio signal with regard to rhythm information by using an autocorrelation function with a sub-band-wise post-processing of the rhythm raw-information; and
- 10
- Fig. 5 a detailed block diagram of means for post-processing of Fig. 4.
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Detailed Description of preferred Embodiments

- 20 Fig. 1 shows a block diagram of an apparatus for analyzing an audio signal with regard to rhythm information. The audio signal is fed via input 100 to means 102 for dividing the audio signal into at least two sub-band signals 104a and 104b. Every sub-band signal 104a, 104b is fed into
- 25 means 106a and 106b, respectively, for examining it with regard to periodicities in the sub-band signal, to obtain rhythm raw-information 108a and 108b, respectively, for every sub-band signal. The rhythm raw-information will then be fed into means 110a, 110b for evaluating the quality of
- 30 the periodicity of each of the at least two sub-band signals, to obtain a significance measure 112a, 112b for each of the at least two sub-band signals. Both the rhythm raw-information 108a, 108b as well as the significance measures 112a, 112b will be fed to means 114 for
- 35 establishing the rhythm information of the audio signal. When establishing the rhythm information of the audio signal, means 114 considers significance measures 112a,

112b for the sub-band signals as well as the rhythm raw-information 108a, 108b of at least one sub-band signal.

If means 110a for quality evaluation has, for example,
5 determined that no particular periodicity is present in the sub-band signal 104a, the significance measure 112a will be very small, and equal to 0, respectively. In this case, means 114 for establishing rhythm information determines that the significance measure 112a is equal to 0, so that
10 the rhythm raw-information 108a of the sub-band signal 104 will no longer have to be considered at all when establishing the rhythm information of the audio signal. The rhythm information of the audio signal will then be determined only and exclusively on the basis of the rhythm
15 raw-information 108b of the sub-band signal 104b.

In the following, reference will be made to Fig. 2 with regard to a special embodiment of the apparatus of Fig. 1. A common analysis filterbank can be used as means 102 for
20 dividing the audio signal, which provides a user-selectable number of sub-band signals on the output side. Every sub-band signal will then be subjected to the processing of means 106a, 106b and 106c, respectively, whereupon significance measures of every rhythm raw-information will
25 be established by means 110a to 110c. In the preferred embodiment illustrated in Fig. 2, means 114 comprises means 114a for calculating weighting factors for every sub-band signal based on the significance measure for this sub-band signal and optionally also of the other sub-band signals.
30 Then, in means 114b, weighting of the rhythm raw-information 108a to 108c takes place with the weighting factor for this sub-band signal, whereupon then, also in means 114b, the weighted rhythm raw-information will be combined, such as summed up, to obtain the rhythm
35 information of the audio signal at the tempo output 116.

Thus, the inventive concept is as follows. After evaluating the rhythmic information of the individual bands, which

can, for example, take place by envelope forming, smoothing, differentiating, limiting to positive values and forming the autocorrelation functions (means 106a to 106c), an evaluation of the significance and the quality,

5 respectively, of these intermediate results takes place in means 110a to 110c. This is obtained with the help of an evaluation function, which evaluates the reliability of the respective individual results with a significance measure. A weighting factor is derived from the significance
10 measures of all sub-band signals for every band for the extraction of the rhythm information. The total result of the rhythm extraction will then be obtained in means 114b by combining the bandwidth individual results under consideration of their respective weighting factors.

15 As a result, an algorithm for rhythm analysis implemented in such a way shows a good capacity to reliably find rhythmical information in a signal, even under unfavorable conditions. Thus, the inventive concept is distinguished by
20 a high robustness.

In a preferred embodiment, the rhythm raw-information 108a, 108b, 108c, which represent the periodicity of the respective sub-band signal, are determined via an
25 autocorrelation function. In this case, it is preferred to determine the significance measure, by dividing a maximum of the autocorrelation function by an average of the autocorrelation function, and then subtracting the value 1. It should be noted that every autocorrelation function
30 always provides a local maximum at a lag of 0, which represents the energy of the signal. This maximum should not be considered, so that the quality determination is not corrupted.

35 Further, the autocorrelation function should merely be considered in a certain tempo range, i.e. from a maximum lag, which corresponds to the smallest interesting tempo to

a minimum lag, which corresponds to the highest interesting tempo. A typical tempo range is between 60 bpm and 200 bpm.

Alternatively, the relationship between the arithmetic
5 average of the autocorrelation function in the interesting
tempo range and the geometrical average of the
autocorrelation function in the interesting tempo range can
be determined as significance measure. It is known, that
the geometrical average of the autocorrelation function and
10 the arithmetical average of the autocorrelation function
are equal, when all values of the autocorrelation function
are equal, i.e. when the autocorrelation function has a
flat signal form. In this case, the significance measure
would have a value equal to 1, which means that the rhythm
15 raw-information is not significant.

In the case of a system autocorrelation function with
strong peaks, the ratio of arithmetic average to geometric
average would be more than 1, which means that the
20 autocorrelation function has good rhythm information. The
smaller the ratio between arithmetic average and
geometrical average becomes, the flatter is the
autocorrelation function and the lesser periodicities it
contains, which means that the rhythm information of this
25 sub-band signal is less significant, i.e. will have a
lesser quality, which will be expressed in a lower and a
weighting factor of 0, respectively.

With regard to the weighting factors, several possibilities
30 exist. A relative weighting is preferred, such that all
weighting factors of all sub-band signals add up to 1, i.e.
that the weighting factor of a band is determined as the
significance value of this band divided by the sum of all
significance values. In this case, a relative weighting is
35 performed prior to the up summation of the weighted rhythm
raw-information, to obtain the rhythm information of the
audio signal.

As it has already been described, it is preferred to perform the evaluation of the rhythm information by using an autocorrelation function. This case is illustrated in Fig. 4. The audio signal will be fed to means 102 for
5 dividing the audio signal into sub-band signals 104a and 104b via the audio signal input 100. Every sub-band signal will then be examined in means 106a and 106b, respectively, as it has been explained, by using an autocorrelation function, to establish the periodicity of the sub-band
10 signal. Then, the rhythm raw-information 108a, 108b is present at the output of means 106a, 106b, respectively. It will be fed into means 118a and 118b, respectively, to post-process the rhythm raw-information output by means 116a via the autocorrelation function. Thereby, it is
15 insured, among other things, that the ambiguities of the autocorrelation function, i.e. that signal peaks occur also at integer pluralities of the lags, will be eliminated sub-band-wise, to obtain post-processed rhythm raw-information 120a and 120b, respectively.

20 This has the advantage that the ambiguities of the autocorrelation functions, i.e. the rhythm raw-information 108a, 108b are already eliminated sub-band-wise, and not only, as in the prior art, after the summation of the
25 individual autocorrelation functions. Above that, the single band-wise elimination of the ambiguities in the autocorrelation functions by means 118a, 118b enables that the rhythm raw-information of the sub-band signals can be handled independent of another. They can, for example, be
30 subjected to a quality evaluation via means 110a for the rhythm raw-information 108a or via means 110b for the rhythm raw-information 108b.

As illustrated by the dotted lines in Fig. 4, the quality
35 evaluation can also take place with regard to post-process rhythm raw-information, wherein this last possibility is preferred, since the quality evaluation based on the post-

processed rhythm raw-information ensures that the quality of information is evaluated, which is no longer ambiguous.

5 Establishing the rhythm information by means 114 will then take place based on the post-processed rhythm information of a channel and preferably also based on the significance measure for this channel.

10 When a quality evaluation is performed based on a rhythm raw-information, which means the signal prior to means 118a, this is advantageous in such, that, when it is determined, that the significance measure equals 0, i.e. that the autocorrelation function has a flat signal form, the post-processing via means 118a can be omitted fully to
15 save computing-time resources.

In the following, reference will be made to Fig. 5, to illustrate a more detailed construction of means 118a or 118b for post-processing rhythm raw-information. First, the
20 sub-band signal, such as 104a, is fed into means 106a for examining the periodicity of the sub-band signal via an autocorrelation function, to obtain rhythm raw-information 108a. To eliminate the ambiguities sub-band-wise, a spread autocorrelation function can be calculated via means 121 as
25 in the prior art, wherein means 128 is disposed to calculate the spread autocorrelation function such that it is spread by an integer plurality of a lag. Means 122 is disposed in this case to subtract this spread autocorrelation function from the original autocorrelation
30 function, i.e. the rhythm raw-information 108a. Particularly, it is preferred to calculate first an autocorrelation function spread to double the size and subtract it then from the rhythm raw-information 108a. Then, in the next step, an autocorrelation function spread
35 by the factor 3 is calculated in means 121 and subtracted again from the result of the previous subtraction, so that gradually all ambiguities will be eliminated from the rhythm raw-information.

Alternatively, or additionally, means 121 can be disposed to calculate an autocorrelation function forged, i.e. spread with a factor smaller 1, by an integer factor, wherein this will be added to the rhythm raw-information by means 122, to also generate portions for lags $t_0/2$, $t_0/3$, etc.

Above that, the spread and forged, respectively, version of the rhythm raw-information 108a can be weighted prior to adding and subtracting, respectively, to also obtain here a flexibility in the sense of a high robustness.

By the method of examining the periodicity of a sub-band signal based on a autocorrelation function, a further improvement can be obtained, when the properties of the autocorrelation function are incorporated and the post-processing is performed by using means 118a or 118b. Thus, a periodic sequence of note beginnings with a distance t_0 does not only generate an ACF-peak at a lag t_0 , but also at $2t_0$, $3t_0$, etc. This will lead to an ambiguity in the tempo detection, i.e. the search for a significant maximum in the autocorrelation function. The ambiguities can be eliminated when versions of the ACF spread by integer factors are subtracted sub-band-wise (weighted) from the output value.

Above that, the compressed versions of the rhythm information 108a can be weighted with a factor unequal one prior to adding, to obtain a flexibility in the sense of high robustness here as well.

Further, there is the problem with the autocorrelation function that it provides no information at $t_0/2$, $t_0/3$... etc., which means at the double or triple of the "base tempo", which will lead to wrong results, particularly, when two instruments, which lie in different sub-bands, define the rhythm of the signal together. This issue is considered by the fact that versions of the autocorrelation

function forged by integer factors are calculated and added to the rhythm raw-information either weighted or unweighted.

- 5 Thus, ACF post-processing takes place sub-band-wise, wherein an autocorrelation function is calculated for at least one sub-band signal and this is combined with extended or spread versions of this function.
- 10 According to another aspect of the present invention, first, the sum autocorrelation function of the sub-bands is generated, whereupon versions of the sum autocorrelation function compressed by integer factors are added, preferably weighted to eliminate the inadequacies of the
- 15 autocorrelation function in the double, triple, etc. tempo.

- According to another aspect, the postprocessing of the sum autocorrelation function is performed to eliminate the ambiguities in the half, the third part, the second part,
- 20 etc. of the tempo, by not just subtracting the versions of the sum autocorrelation function spread by integer factors, but weighting them prior to subtraction with a factor unequal one and preferably smaller than one and larger than zero, and to subtract them only then. Thereby, a more
- 25 robust determination of the rhythm information becomes possible, since unweighted subtracting provides a full elimination of the ACF ambiguities merely for ideal sinusoidal signals.

- 30 While this invention has been described in terms of several preferred embodiments, there are alterations, permutations, and equivalents which fall within the scope of this invention. It should also be noted that there are many alternative ways of implementing the methods and
- 35 compositions of the present invention. It is therefore intended that the following appended claims be interpreted as including all such alterations, permutations, and

equivalents as fall within the true spirit and scope of the present invention.